Claims

1. Method for enhancing the quality of a received acoustic signal, in particular speech signal, wherein the received acoustic signal has been generated by a single microphone, wherein the received acoustic signal is subjected to an analysis of characteristics,

   wherein

   the analysis is used to estimate one or more virtual microphone signals, which are parts of the received acoustic signal,
   and the one or more virtual microphone signals are used to generate an enhanced quality acoustic signal, in particular with reduced echo and/or reduced reverberation compared to the received acoustic signal.

2. Method according to claim 1,

   wherein

   a) the received acoustic signal is subjected to an analysis detecting the time period d1 between direct sound and the onset of reverberation sound within the received acoustic signal,
   b) a delay signal is generated by delaying the received acoustic signal by the time period d1,
   c) a modified delay signal is created by modifying the delay signal applying a set of modification parameters,
   d) a first virtual microphone signal is generated by subtracting the modified delay signal from the received acoustic signal,
   e) the first virtual microphone signal is subjected to an analysis

generating one or several analysis parameters, and
f) the modification parameters are adapted within a feedback loop, optimizing the analysis parameter, in particular minimizing the overall amplitude of the first virtual microphone signal.

3. Method according to claim 2, wherein the enhanced quality acoustic signal is generated by amplifying the level of the first virtual microphone signal, in particular to a normal loudness.

4. Method according to claim 2 for generating an nth virtual microphone signal, with $n \in \mathbb{N}$, $n \geq 2$, wherein
an nth intermediate signal is generated by subtracting the first to (n-1)th virtual microphone signal from the received acoustic signal,
a') the nth intermediate signal is subjected to an analysis detecting the time period dn between the onset of sound and the onset of reverberation sound within the nth intermediate signal,
b') an nth delay signal is generated by delaying the nth intermediate signal by the time period dn,
c') an nth modified delay signal is generated by modifying the nth delay signal applying a set of modification parameters,
d') an nth virtual microphone signal is generated by subtracting the nth modified delay signal from the nth intermediate signal,
e') the nth virtual microphone signal is subjected to an analysis generating one or several analysis parameters, and
f') the modification parameters are adapted within a feedback loop, optimizing the analysis parameter, in particular minimizing the overall amplitude of the nth virtual microphone signal.

5. Method according to claim 4, wherein the enhanced quality acoustic signal is generated by adding a number of N virtual microphone signals, with $N \in \mathbb{N}$, $N \geq 2$, wherein the mth virtual microphone signal is delayed

by a time period tm = $\sum_{i=m}^{N-1} di$, with m $\in$ [1,..., N-1], and the Nth virtual microphone signal is undelayed.

6. Method according to claim 4, wherein the modification in steps c) and/or c') is performed by a finite impulse response unit, and wherein the modified time period of the finite impulse response unit is at least as long as the reverberation time of the received acoustic signal.

7. Method according to claim 4, wherein the determination of the analysis parameters in steps e), and/or e') is performed by a least mean square method and/or a normalized least mean square method.

8. Method according to claim 4, wherein the received acoustic signal and/or the nth intermediate signal and/or the delayed signal and/or the nth delayed signal is/are subjected to a Fourier transformation, and the modification is performed in the frequency domain.

9. Method according to claim 4, wherein in steps a) and/or a') the onset of the reverberating sound in the signal amplitude vs. time diagram of the received acoustic signal and/or nth intermediate signal is determined by observing an edge of the signal amplitude following a time period of substantially constant signal amplitude within a limited frequency interval, in particular within 100-300 Hz.

10. Method according to claim 1, wherein a start of the received acoustic signal is detected, and that the following steps are performed recursively in one or more cycles:

a) observing the stored signal, i.e. in the first cycle the received acoustic signal, else the processed signal derived in the preceding step c) to be

further cleaned, for a signal excitation indicating the start of a disturbing echo and/or reverberation signal;

b) determining the time delay d between the start of the received acoustic signal and the start of the disturbing echo and/or reverberation signal, and estimating the magnitude of the disturbing echo and/or reverberation signal;

c) generating a processed signal by subtracting a compensation signal from the stored signal, wherein the compensation signal is derived from the stored signal by shifting the stored signal by the time delay d and scaling the stored signal with the estimated magnitude,

wherein the processed signal of the last cycle is defined to be the first virtual microphone signal.

11. An acoustic signal quality enhancement device, comprising means for performing a method according to claim 1.

12. A computer terminal comprising an input for a received acoustic signal, in particular a microphone and/or a data carrier device and/or a data line, an output for an enhanced quality acoustic signal, in particular a loudspeaker and/or a data carrier device and/or a data line, and means for performing a method according to claim 1.